CHAPTER 7

Sound Engineering

LEARNING OBJECTIVE

In this chapter, the student can

- Understand the characteristics of sound waves
- Study the PA system and Audio power amplifier circuits
- Understand the acoustic techniques in auditorium
- Study the theater sound system DTS/ DOLBY
- Study the applications of Acoustic Engineering
- Study the effects of noise pollution

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CONTENT

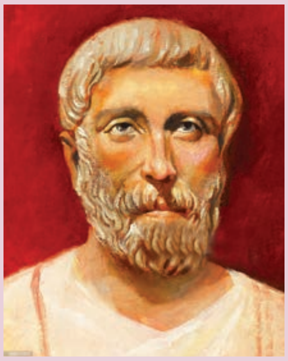
HISTORY OF SOUND

One of the first discoveries regarding sound was made by mathematician Pythagoras, in sixth century BC. He noted the relationship between the length of a vibrating string and the tone it produces.

Italian physicist, Galileo Galilei, was the first scientist to record the relationship between the frequencies of the wave in terms of its pitch it produces. Since the sound waves produced by musical instruments vary in pitch, this was a very significant discovery. In the 1640s, Marin Mersenne was the first to measure the speed of sound in the air.

Robert Boyle discovered in the year 1660 that sound waves must travel in medium and this lead to the concept that sounds is a pressure change.

Galilio Galili



Phythagoras

Robert Boyle

7.1 Introduction

In this chapter, we can learn some fundamental knowledge and skills to enter into the field of sound engineering.

First, we have to understand, what do you mean by sound and audio? Sound is a frequency caused by vibration that can be heard by humans, animals or any device that can pick up those frequencies.

Audio means 'of sound or of the reproduction of sound'. Specifically it refers to the range of frequencies detectable by the human ear, approximately 20 Hz to 20 kHz. The audio work involves the production, recording, manipulation and reproduction of sound waves.

We can also study about the PA system, power amplifier circuits, acoustic application and DTS/DOLBY systems.

DO YOU KNOW?

Sound travels in different mediums in different speed

Speed of sound in solids : 5960 m/s

Speed of sound in liquids : 1482 m/s

Speed of sound in Air (gases) : 334 m/s

Sound cannot travel in Vacuum

7.2 Characteristics of Sound waves

Figure 7.1 shows the waveform of a sound wave. We know that sound travels in the form of wave. A wave is a vibratory disturbance in a medium which carries energy from one point to another without having direct contact between the two points.

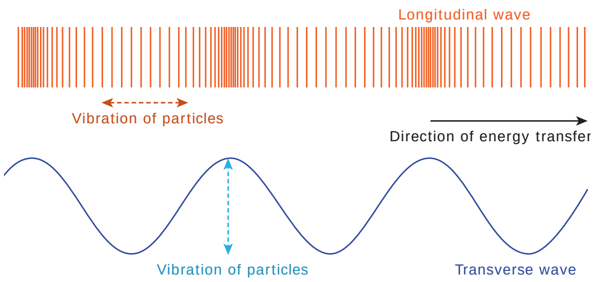


FIGURE 7.1 Longitudinal and Transverse Wave

There are two types of waves:

1. Longitudinal waves
2. Transverse waves.

Longitudinal Waves: A wave in which the particles of the medium vibrate back and forth in the 'same direction' in which the wave is moving. Medium can be solid, liquid or gases. Therefore, sound waves are longitudinal waves.

Transverse Waves: A wave in which the particles of the medium vibrate up and down 'at right angles' to the direction in which the wave is moving. These waves are produced only in a solids and liquids but not in gases.

Sound is a longitudinal wave which consists of compressions and rarefactions travelling through a medium.

Sound wave can be described by five parameters as shown in Figure 7.2: They are

1. Wavelength
2. Amplitude
3. Time-Period
4. Frequency
5. Speed or Velocity

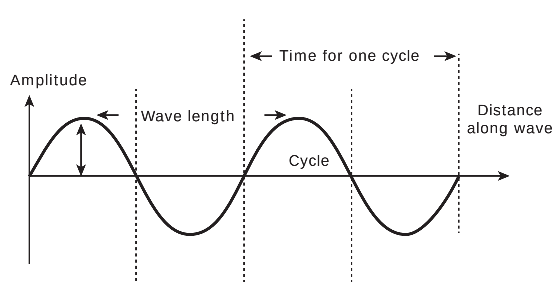


FIGURE 7.2 Characteristics of Sound waves

1. Wavelength

The minimum distance in which a sound wave repeats itself is called its wavelength, i.e., it is the length of one complete wave. It is denoted by a Greek letter λ (lambda). In a sound wave, the combined length of a compression and an adjacent rarefaction is called its wavelength. Also, the distance between the centers of two consecutive compressions or two consecutive rarefactions is equal to its wavelength. The S.I unit for measuring wavelength is metre (m).

2. Amplitude

The Maximum extent of a vibration or displacement of a sinusodal oscillation, measured from the position of equilibrium. Amplitude is the maximum absolute valued of a periodically varying quantity In fact the amplitude is used to describe the size of the wave. The S.I unit of measurement of amplitude is meter (m). The amplitude of the vibrating body producing the sound determines the loudness of the sound. If the amplitude is higher, the sound produced is louder.

3. Time-Period

The time required to produce one complete wave or cycle is called time-period of the wave. Now, one complete wave is produced by one full vibration of the vibrating body. So, we can say that the time taken to complete one vibration is known as time-period. It is denoted by letter T. The unit of measuring the time-period is second (s).

4. Frequency / pitch

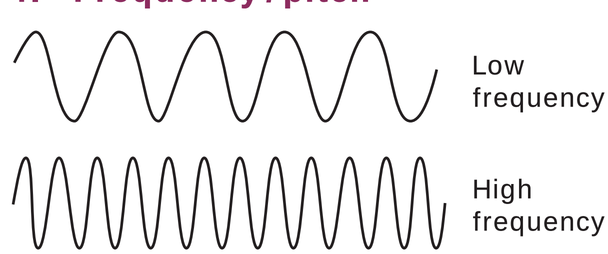


FIGURE 7.3 High and low frequency

The number of complete waves or cycles produced in one second is called frequency of the wave. Since one complete wave is produced by one full vibration of the vibrating body, so we can say that the number of vibrations per second is called frequency. The S.I unit of frequency is Hertz or Hz. The pitch of a sound is the ear and brain interpreting the frequency of the sound. When there is a high frequency, the ear interprets the sound as a higher pitch, when the frequency is low, the ear hears the sound as a low pitch. It is a measure of sound in frequency and is shown in Figure 7.3.

5. Speed or Velocity

The distance travelled by a wave in one second is called velocity of the wave or speed of the wave. It is represented by the letter v. The S.I unit for measuring the velocity is meters per second (m/s or ms^{-1}).

7.3 Microphones

7.3.1 Lavalier Microphone (collar Mic)

Lavalier Microphone is also known as lav, a lapel or lap microphone is shown in Figure 7.4. It is a very small condenser mic designed to pick up speech from a single person. This mic is widely used for TV program, Public Address systems etc. Lavalier mic is usually attached to the subject's clothing with a specialized clip. Obviously the preferred position on the lapel or thereabouts. This provides consistent close range sound pickup and ideal for interview situations in which each participant have their own mic. It also means the subject do not worry about the mic techniques.

Further, the cable can be discreetly hide under the clothing. If there is nowhere to place the mic on the subject's chest, it can be fixed on the collar. Lavalier mic can be quite susceptible to noise caused by movement of the subject position, i.e., it cannot be moved around too much, and make sure that the cable cannot be pulled by anyway. A small wind filter can be used to reduce wind noise.

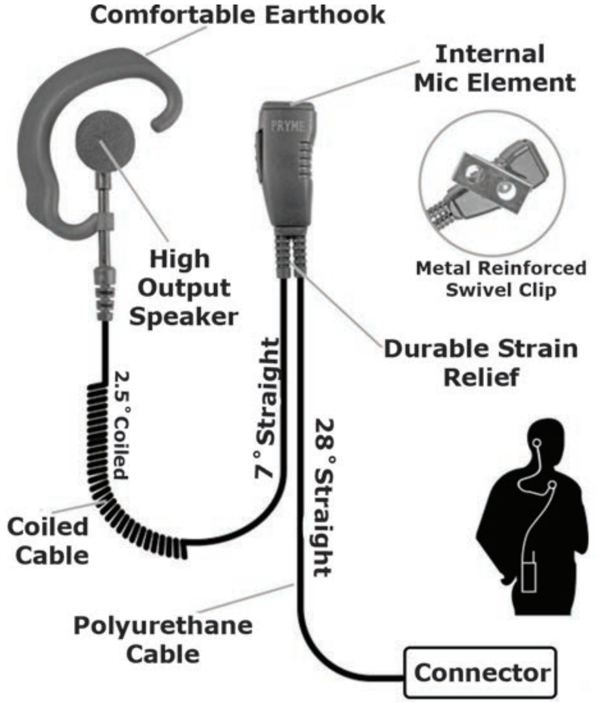


FIGURE 7.4 Lavalier Microphone

7.3.2 Crystal microphone

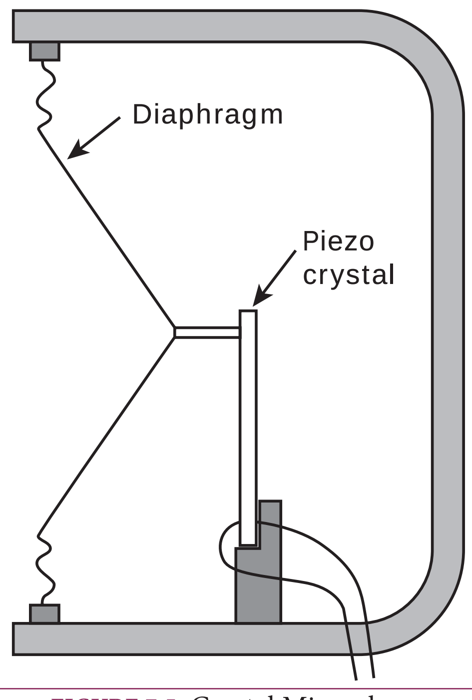


FIGURE 7.5 Crystal Microphone

Crystal microphone or ceramic microphone as shown in Figure 7.5 is generally a low cost microphone providing a high output voltage in the order of 10 to 100 mV. The main use of crystal microphone technology is within transducers used for a variety of monitoring applications and for automotive transmitters / sensors. This type of microphone is working on the principle of piezoelectric technology. The piezoelectric effect is the ability of certain materials to generate an electric charge when mechanical stress is applied.

In this type of microphone, alternating voltage is produced when sound makes the diaphragm vibrates. The impedance is very high usually in the order of 1 to 5 Mega Ohms. The charge produced by the piezoelectric action of the crystal is converted into voltage using electronic circuits. The natural crystals used in this type of microphones are Rochelle salt and quartz.

Crystals which demonstrate the piezoelectric effect produce voltages when they are deformed. The crystal microphone uses a thin strip of piezoelectric material attached to the diaphragm. The two sides of the crystal acquire opposite charges, when the crystal is deflected by the diaphragm. The charges are proportional to the amount of deformation and disappear, when the stress on the crystal disappears. Early crystal microphones used Rochelle salt, because of its high output, but it was sensitive to moisture and somewhat fragile. Later, microphones used ceramic materials such as barium Titanate and lead Zirconate Titanate.

7.3.3 MicroElectroMechanical Michrophones (MEMS)

MEMS microphones are extremely small microphones designed to fit on a silicon chip. They are based on the same working principles as condenser microphones. They have an analog-to-digital converter (ADC) module integrated on the same chips. It converts the analog input into digital values, which are used by the modern electronic devices. MEMS find applications in modern-day electronic gadgets, such as cellphones, tablets, laptops, automotive industry, etc. Figure 7.6 shows MEMS microphones.

7.4 Head phones

Headphones are a pair of small speakers, which are used for listening to sound from a music player, computer, Laptop, Smart phone or such other electronic devices. It is also called as earphones or Headset. The modern headphones are available in much smaller format, which can be inserted into

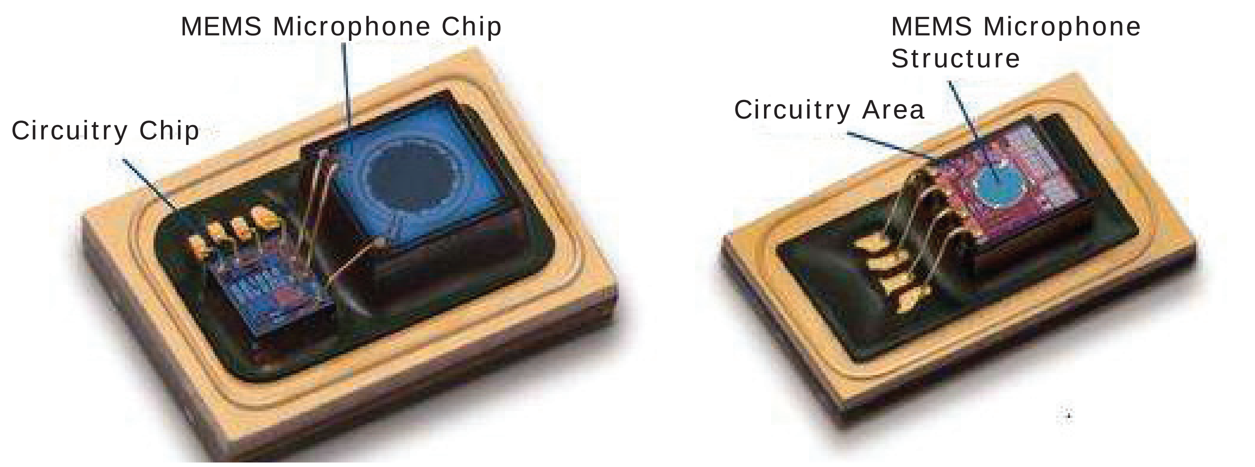


FIGURE 7.6 MicroElectroMechanical Michrophones (MEMS)

the ear and are commonly called ear buds. Nowadays, headphones can be either wireless or wired.

> **DO YOU KNOW?** **The first headphone was developed in 1910 by the US navy. It was simple and was used as an earpiece device without complicated electronics.**

Working functions

Headphone works like a speaker and opposite to microphone as shown in Figure 7.7. It converts the electrical signal into the sound signal through the vibration of the magnet, thereby vibrating the surrounding air particles.

Once the electrical signal makes its way through the wires into the headphones, it reaches a driver unit.

There are three types of driver units. They are

1. Dynamic driver
2. Planar magnetic driver
3. Electrostatic driver

Most of the headphones uses Dynamic driver unit. The Dynamic driver unit uses three main parts to work. They are

1. Permanent magnet
2. Electromagnetic coil
3. Diaphragm

Each ear cup has one permanent magnet, which firmly in place and the other is an electromagnet that moves. When the electrical signal hits the ear cup, it sent to the electromagnet, which rapidly switches its polarity back and forth depending on the

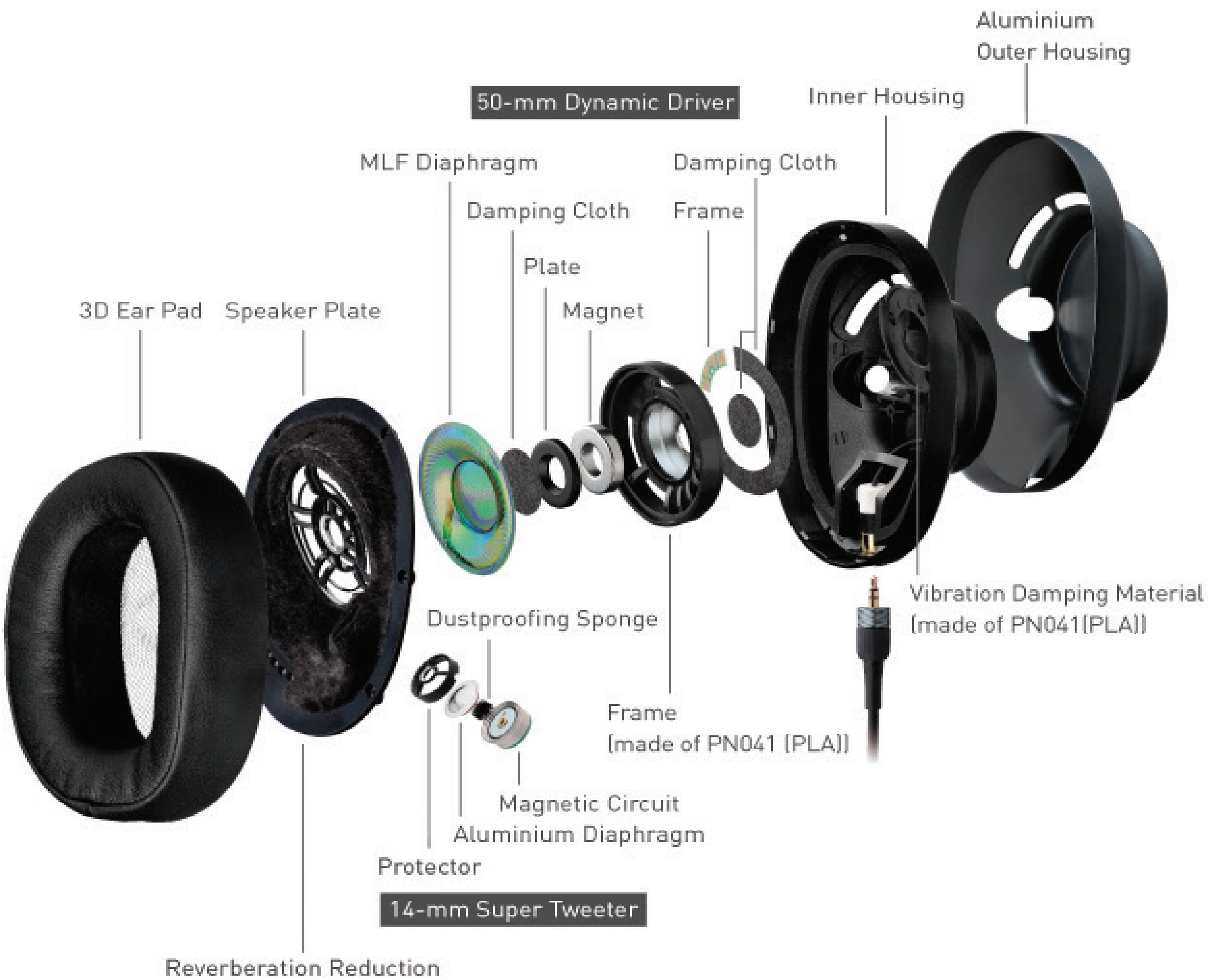


FIGURE 7.7 Headphones

pattern it sent or the sound being reproduced. When the electromagnet switches, its polarity rapidly repelled and attracted towards the permanent magnet, which makes it vibrates. Those vibrating electromagnets are attached to what is called a 'diaphragm', which is a thin membrane. When the electromagnet vibrates the diaphragm, which causes the air around it to vibrate, this is what we called sound. Different frequencies vibrate at different rates so the electromagnet vibrates faster to produce high tones, or slower to produce slow tones. When we turn the volume up or down, the vibrations are more or less intense, which causes the air to vibrate more or less.

7.5 Loud Speakers

7.5.1 Flat panel speakers

There are several kinds of flat panel speakers. Engineers have been working on flat speakers for many decades so as to decrease the size of speaker boxes. The standard flat panel speaker has an exciter attached to a square panel. The flat panel acts as a diaphragm. Different materials can be used as a diaphragm such as Vinyl or Styrofoam.

The standard flat panel electro dynamic loud speaker has been difficult to make because, it is difficult to vibrate the entire flat surface evenly while creating good frequency response, thus other speaker types have evolved to make a speaker in a flat form. Figure 7.8 shows a flat panel loud speaker.

FIGURE 7.8 Flat panel speakers

Types of flat panel speakers

1. Ribbon
2. Planar magnetic
3. Electrostatic

7.5.2 Piezoelectric Speaker

Figure 7.9 shows Piezoelectric Speakers. Piezoelectric Speakers use an expanding and contracting crystals to vibrate the air and produce sound. This type of speakers are limited in frequency response, therefore they are only used as tweeters or in small electrical devices like watches/ clocks to make simple sounds. It may be possible in the future that the technology may improve, by the way of providing a speaker with good sound charecteristics and durability.

FIGURE 7.9 Piezoelectric Speakers

Piezoelectric are solid state technology which makes them durable and good for use as a microphone under water. These speakers are used as michrophones in submarines warefare, they can detect other microphones and hear sounds of other vessels.

7.6 Acoustics Engineering

Acoustic is a branch of physics concerned with the study of sound (mechanical waves in gases, liquids and solids). Acoustics have many application in the everyday world and this technology is called acoustical engineering.

The study of acoustics can be sub divided into three parts. They are production, transmission and reception. All these elements are necessary for sound generation and reproduction. For example, a ringing alarm clock cannot be heard, if it is placed inside a vacuum container. Without air, sound produced by the clock has no medium through which it can travel.

Application of Acoustics

Acoustics have wide range of applications in many fields. We discuss some of the important applications in this Section.

1. Noise and environmental Acoustics

Noise Specialists are mostly concerned with making our world a (quitter) peaceful place. They study man-made noise caused by machinery, transportation using roadways, railways, aircraft and general activities. Knowledge produced by these scientists can be used to redesign noisy machinery or to recommend ways of redesign noiseless machinery or to recommend ways of shielding the noise. They also help law makers and public officials to create rules for limiting exposure of noise.

2. Medical Acoustics

Medical researchers and Doctors used acoustics to study, diagnose and treat different types of ailments. The study of material acoustics includes the use of ultrasound and other acoustical techniques to learn how different types of sound interact with cells, tissues, organs and entire organisms. Biomedical acoustians may work with engineers, physician and speech therapist.

3. Musical Acoustics

Musical acoustians study the science of how music is made, travel and heard. Since musical acoustics combines elements of art and science, people with training in this field can work in the entertainment industry and much more.

4. Speech and Hearing Acoustics

Hearing specialist and speech scientists are interested in how our ears sense sounds and what types of sounds can damage our ears, how speech is made, travel and heard. People interesting in hearing and speech come from many different fields, including physics, speech and hearing science, experimental psychology, linguistics, electrical engineering and others.

5. Architectural Acoustics

Architectural acoustians study how to design buildings and other spaces that have pleasing sound quality and safe sound levels. Architectural acoustics include the design of concert halls, classrooms and even heating systems, where they work with musical acoustians and noise specialists.

7.7 Acoustics in Auditorium and Theater

A most important part of the auditorium design is the acoustics. We start with a brief description of how your ear works in the context of listening.

FIGURE 7.10 Auditorium

How the ear works

The human ear has developed over the evolution of humans into an organ capable of receiving the short term fluctuations of air pressure around us and extracting vast amounts of information from them. These short term air pressure fluctuations are commonly called sound waves.

When listening in an auditorium, human brain tries to make sense of the cacophony of sound waves arriving at the ears. Here, it is useful to think of the concept of the flicker fusion threshold.

When the ear is presented with reflections of a sound that arrive much later than the direct sound, the brain interprets those as echoes, and is able to separate them from the original sound. Once the reflections arrive soon enough, after the direct sound to pass the threshold of 50 milliseconds, the brain is then able to fuse the reflected energy with the direct sound and use it to enhance the intelligibility of the speech being heard.

Acoustic design principles

The main driver behind acoustic design in auditoriums derived from the phenomenon described above. Usually, keep and enhance 'early' reflections to arrive at the listener not more than 50 milliseconds after the direct sound. Then, dampen or reduce the 'late' reflections that would arrive at the listener more than 50 ms after the direct sound. At a given listener location, if there is more early acoustic energy than late, speech will be intelligible. To that end, surfaces should be provided and shaped to provide such early reflections, and reflection paths that provide late acoustic energy should be made acoustically absorptive. This leads to certain rules of thumb as summarized below:

1. Shoebox-shaped rooms provide for strong early lateral reflections (even more important for music, but quite helpful for speech as well)
2. Reflections down from a ceiling can often provide early reflections, and therefore should be made acoustically hard (reflective)
3. The back walls of an auditorium have a risk of providing late reflections – both to the audience and to the stage: Providing acoustic absorption at such locations is usually helpful. This could be in the form of fabric panels, slatted wood finish, acoustic plaster or even acoustic drywall.
4. The audience seats and the audience themselves are usually the biggest acoustic absorption in the room. The use of the right amount of acoustic absorption in the seats can serve as a great way to achieve the acoustic goals of the space.

7.8 Audio Power Amplifier - Types

7.8.1 Audio Amplifier using TBA 810 IC

Figure 7.11 shows the circuit and pin details of audio amplifier using TBA 810 IC. It is a simple, cost-effective and capable of producing 7-watts output audio amplifier. The amplifier is fabricated as monolithic integrated circuit in a 12-lead quad–in–line plastic package, intended for use as a low frequency class B amplifier. The circuit is used in low-power audio amplifier designs.

Construction and Working Functions

The circuit shown in Figure7.11 is constructed with TBA 810 IC and few RC components. The voltage requirement of the IC is 6 V – 20 V (500 mA) and drives 4Ω to 16Ω speaker at output.

The audio signal input is given to the pin no.8 of IC through volume control $VR_{1.}$ The amplified audio output is taken from pin

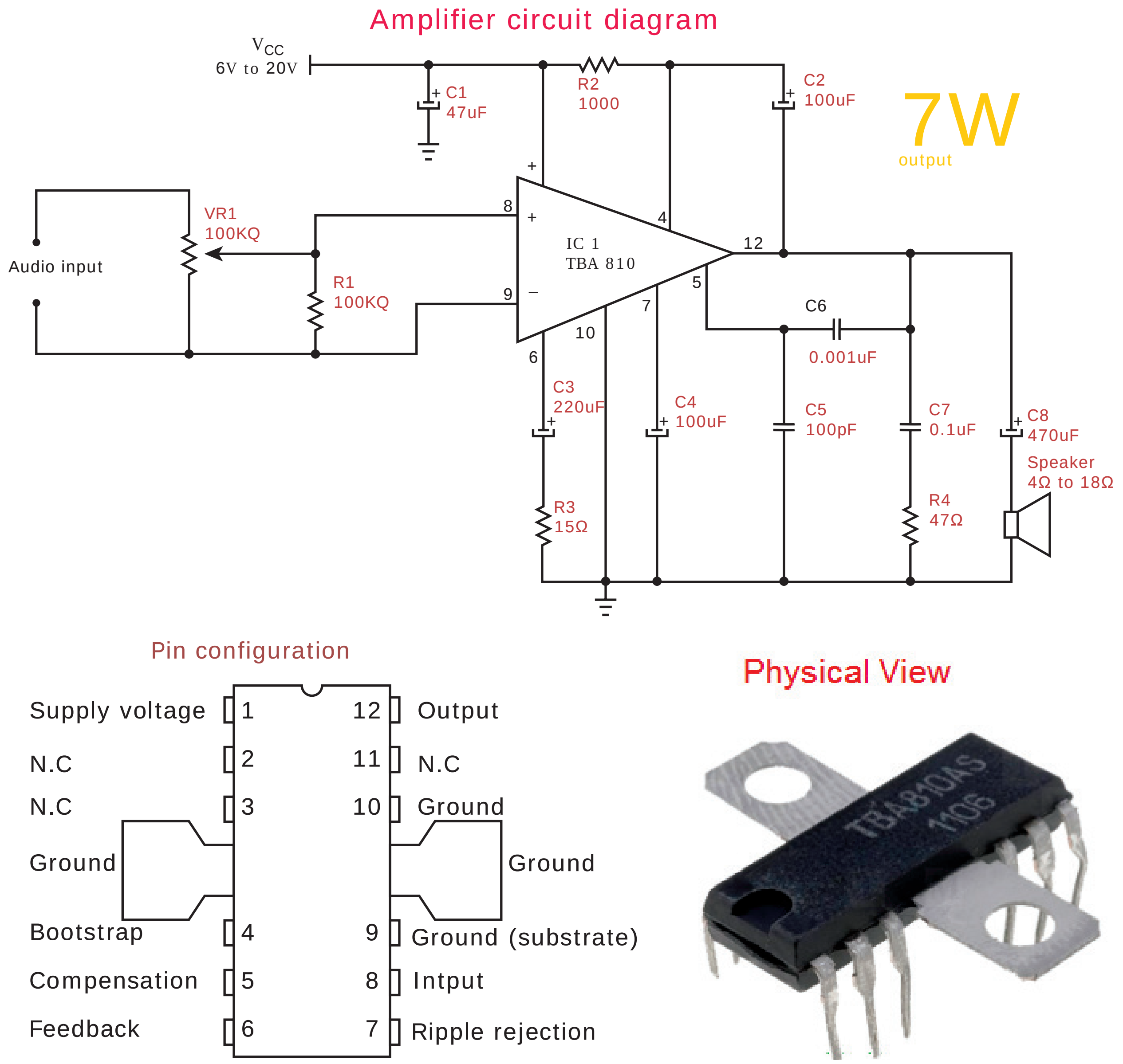


FIGURE 7.11 Audio Amplifier using TBA 810 IC

no.12 and given to the speaker. The IC can be covered with heat sink and the tapper on both sides must be grounded. If the supply voltage is between 4 V – 6 V, the circuit provides a low power output (1 Watt). For 6 V - 20 V, the audio output power increases to higher level (7 Watts).

7.8.2 Audio amplifier using LA4440 IC

Figure 7.12 shows the audio amplifier using LA 4440 IC. This IC is most suitable for low power audio applications. The amplifier circuit has good ripple rejections (46 dB) and good channel separation. The IC is a dual channel audio amplifier with low distortion over a wide range from low frequencies to high frequencies. It is build with heat sink as a thermal protector for better performance. LA 4440 has over voltage, surge voltage protector and pin-to-pin short protector. These specific features are making the LA4440 as a unique audio amplifier.

Construction and Working Functions

This circuit is designed to provide stereo amplification to the input audio signal,

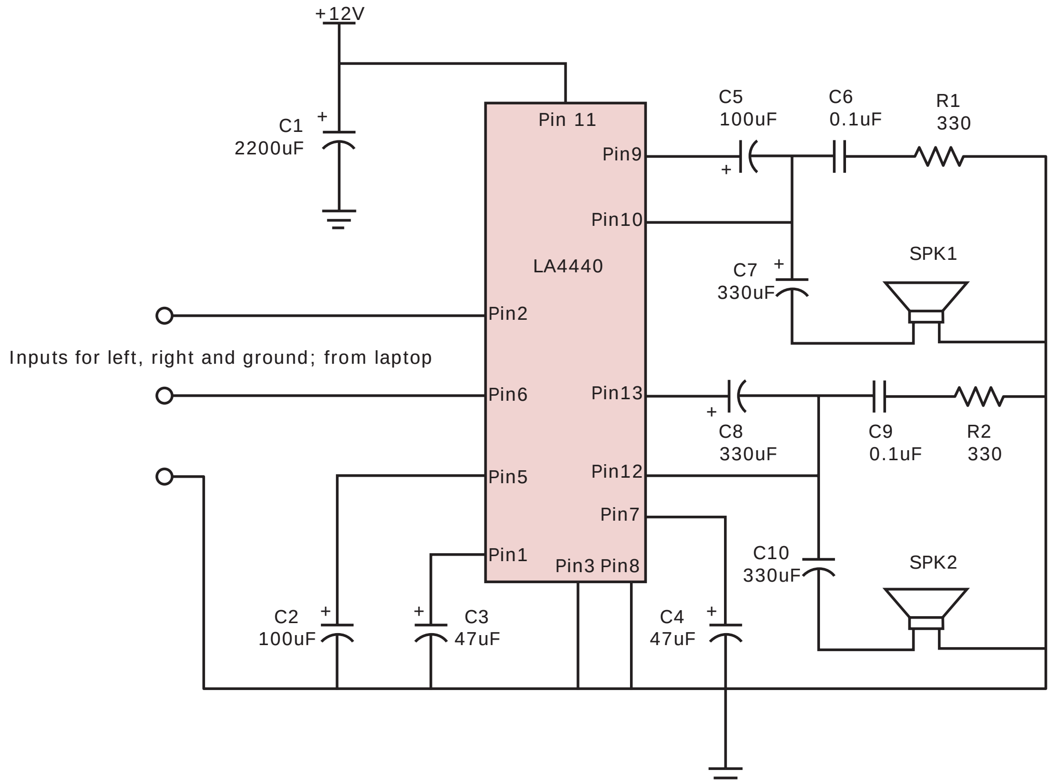


FIGURE 7.12 The Audio amplifier circuit diagram

pin numbers 2 and 6 takes audio input signals. The amplified output audio signal is taken out from pin numbers 10 and 12. Maximum supply voltage to this amplifier is +18 V and it operates in the temperature range of -20° to +75 °C.

The common supply voltage is +12V. This amplifier gives 30 k-ohms input resistance. By adding input variable resistor, we can control output volume.

Activity

Assemble the circuit given in figure 7.11 and 7.12 using PCB

7.8.3 Audio Amplifier using TDA 2003

Figure 7.13 shows the circuit of the audio amplifier using TDA 2003. The IC is a monolithic, which contains a preamplifier, driver amplifier and output amplifier. The amplifier has very low harmonic distortion and high output current capability.

Construction and Working Functions

The circuit is constructed with the TDA 2003 IC, which has only 5-pins and all are function pins. The IC has built-in over temperature protection and short circuit protection features.

The audio input is given to the pin 1 (non-inverting pin) of the IC. Pin 2 (inverting pin) is connected with capacitor C_4 and voltage divider resistors R_2, R_3, which acts as a feedback path. The loud speaker is connected between pin 4 and pin 3 (GND). The supply voltage is (6V – 12V) given to pin 5 and pin 3 is grounded. Capacitors C_1 and C_2 are used to filter out the power supply fluctuations. This circuit provides a 10 watts output.

7.9 Audio effects

Mono and stereo are two classifications of reproduced sound. The main difference between mono and stereo comes with the number of audio channels used in each.

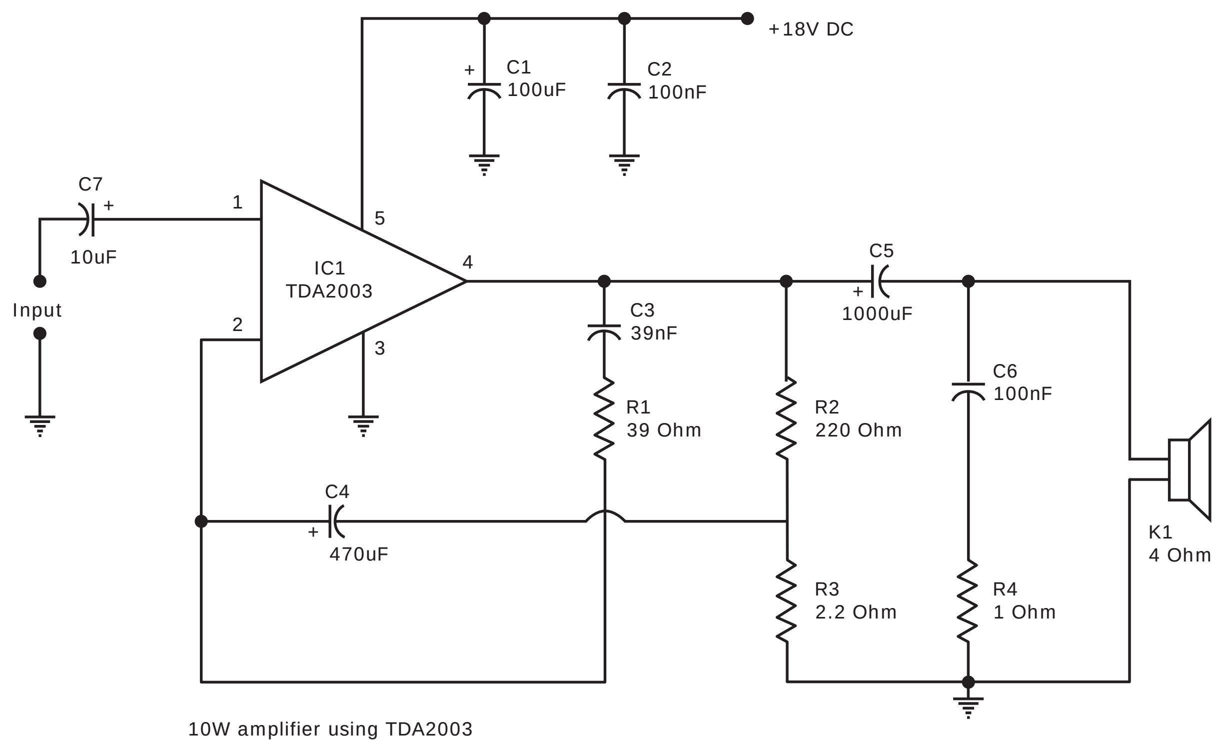


FIGURE 7.13 Audio Amplifier using TDA 2003

Mono is the term used to describe the sound that is only from one channel, while stereo uses 2 or more channels to provide an experience much like being in the same room where the sound is created.

7.9.1 Monaural sounds

Mono is a short version on monaural sound, having only one source for the audio. From the Figure 7.14, we can

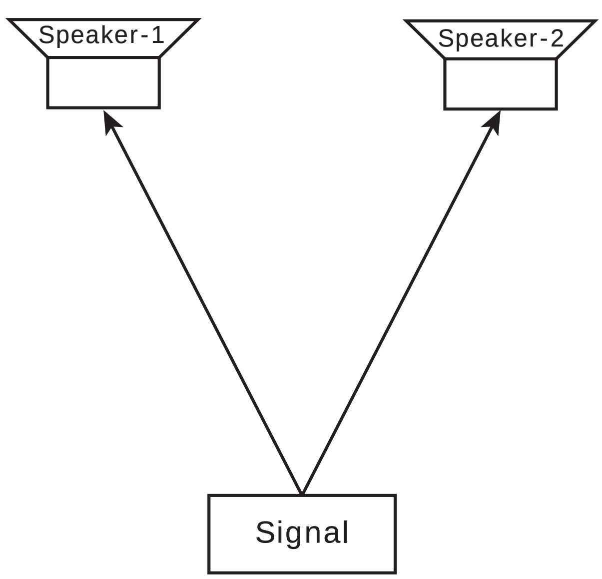


FIGURE 7.14 Monaural sounds

understand that the sound comes from only one sources even though it is transported to two different speakers, i.e., the content of the signal is always the same. When listening to music or other auditory speeches using headphones, we cannot hear any difference by removing one earphone.

Mono is still widely used in situations where stereo only takes up bandwidth and offers no advantages. A good example for this is in voice communications like in talk radio and telephone calls. The equipment needed to record mono sound is only a single microphone and the data it acquires is automatically stored in magnetic tape or converted to digital formats for storage.

7.9.2 Stereophonic sounds

Stereo is a short version of stereophonic sounds. In stereo, several channels are used to transport audio signals to speakers and thus to a listeners' ears. Generally, stereo uses two channels, but it can use more. In the most common set up, one channel is transported to one speaker and the other channel to another speaker.

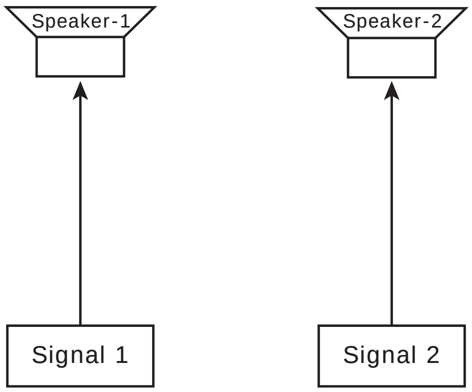


FIGURE 7.15 stereo phonic sounds

Figure 7.15 shows the usual setup for stereo sounds. There are two different sources that send their individual signal to one speaker each. In this system, the sounds that are transported entirely to the right speaker will appear to come from a listener's right side. The signal not only transported to one speaker in it's entirely through, but it is transported proportionally as well. That is, a small proportion of the sound can be transported to the right speaker, while the rest is sent to the left one creating more 3-dimensional hearing experiences. Sounds that are equally transported to both the speakers appear to come from the center.

This is all based on the typical set up of two sources of sound that are transported to the two speakers. Thus, stereo is used to create an inspiration of sounds coming from different directions as well as setting the sound in perspective to one another and the listener. This is especially useful in movies and audio plays to emerge the listener/viewer into the story. It is also used in music. Particularly in film songs, the guitar part is send to one speaker, while the bass is send to the other. Headphone users are easily identifying the stereophonic sounds. Removing one earphone can reveal that a particular instrument or sound is only transported to either the left or the right ear.

7.9.3 Equaliser

It is a control used for boosting or reducing (attenuating) the levels of different frequencies in a signal. We have the experience of hearing the treble / bass control on public address amplifier and home audio equipment, which is nothing but a basic type of equalizer. The treble control adjusts the high frequencies whereas the bass control adjusts the low frequencies. This is adequate for very basic adjustments, i.e., it only provides two controls for the entire frequency spectrum, so each control adjusts a wide range of frequencies.

Advanced equalizations system provides a fine level of frequency control. They enable to adjust a narrower range of frequencies without affecting neighboring frequencies. Equalizer is the most commonly used unnatural sound system. For example, if a sound was recorded in a room which accentuates high frequencies, an equalizer can reduce those frequencies to a more normal level. It can also be used for applications such as making sounds more by reducing the feedback.

7.9.4 Ambience

Ambience and ambient sound generally denotes the surrounding sounds that are present in a scene or location, such as wind, water, birds, forest murmurs, electrical hum, room tone, office clatters, traffic, and neighborhood mutterings. Ambient sound can provide a specific atmosphere of a public site in the construction of the diegetic space or the interior world of a film or sound-based media network. To the sound artist and practitioners, ambient sound injects life and substance not only to what we see on the cinematic screen but, also to the off-screen story world. The practitioners use the material layers of ambient sound to construct the experience of presence.

7.10 PA System

Figure 7.16 shows the block diagram of public address system. The functions of different blocks are as follows.

Microphone

It converts sound to an equivalent electrical signal. Generally, two or three microphone can be connected with one auxiliary input for CD is also provided.

Mixer

The output of microphone is fed to the mixer stage. The mixer stage is used to isolate different channels from each other before they are fed to the amplifier.

Voltage Amplifier & Processing Circuits

The voltage amplifier is used to amplify the mixer output further. The processing circuit block consists of the 'master gain control' and the 'tone control circuits'. The tone control circuits consist of the bass and treble control circuits. The bass control circuit will amplify or cut the low frequency signals and the treble control will amplify or cut the high frequency signals.

Driver and Power Amplifier

The driver amplifier drives the power amplifier to give more power. It is basically a voltage amplifier. The power amplifier gives the desired power amplification to the input signal. The push-pull type of amplifier is generally used because this type eliminates the even harmonics from the output of the amplifier and avoids the core saturation of the output transformer. The power amplifier drives the loud speakers. Matching transformers are used between them to match the low speaker impedance to the output impedance of the power amplifier.

Requirements of PA system

1. It must avoid the acoustic feed Back
2. Distribute the sound intensity uniformly
3. Reduce reverberations
4. It must use proper speaker orientations.
5. Select proper microphones and loudspeakers .
6. It should create a sense of direction
7. Loud speaker impedance should be matched properly
8. Proper grounding should be provided
9. Use closed ring connection for loudspeakers

7.11 Theater Sound system- DTS & DOLBY

Just like music, surround sound format comes in many standards. The two most popular ones supported by a broad range of high-end audio systems such as DTS and Dolby Digital.

DTS is the abbreviations of digital theater system, a popular home theater audio format that was developed in 1993. Dolby Digital is the name for audio compression technology developed by the Dolby Labs. Both systems are for the

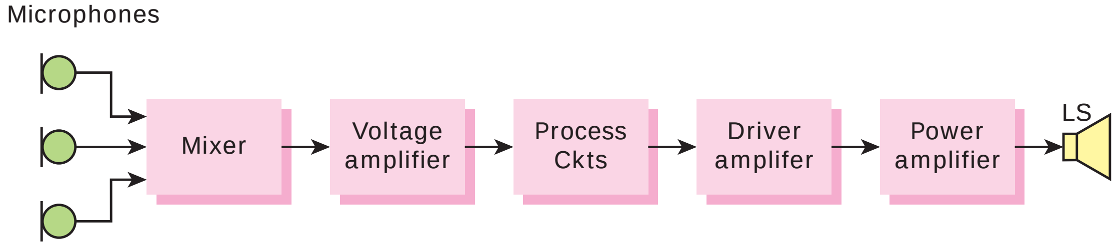


FIGURE 7.16 PA system

development of surround sound audio technology in movie production.

These systems provide sound codes for 5.1, 6.1 and 7.1 setups, where the first number represents the number of surround speakers, and the '1' is a separate channel for subwoofer.

Both formats utilize 'perceptual' data reduction techniques to remove useless data in PCM signal output, thereby processing high fidelity sound. In addition to the 5.1 to 7.1 speaker playback, different formats offer cutting edge audio technology geared towards enhancing the sound quality. For instance, DTS and Dolby digital use compression to same space either on the disc, as is the case with Blue Ray and DVDs or on streaming bandwidth for services like NETFLIX.

Some versions of Dolby Digital and DTS are 'lossy' which means they have a degree of audio degradation from the original source, while others are lossless.

Dolby, for example, has a lossless version, Dolby True HD, and a lossy version table up very little space on Blue–Ray disc. DTS also has a lossless version, DTS –HD master Audio, that supports 7.1 channels speaker setup.

7.12 Audio Recording

Audio recording techniques have developed dramatically in recent years. Excellent digital equipment with vast capabilities is now quite affordable. Low cost and high technology has meant that many people are leaping directly to sophisticated recording equipment for their first recording experience.

7.12.1 Basic Recording / Multitrack Recording

The recording process, whether accomplished with a cassette recorder, digital multi-track recorder, hard disk recorder or any other recording medium, is essentially the same. The goal is to capture sounds onto a master recording. To do this, recording engineers employ a two-step system:

1. **Multitrack Recording** - the process of recording and overdubbing various instruments and vocals, each to its own "track."
2. **Multitrack Mixdown** - the process of simultaneously re-recording these multiple tracks down to one set of stereo tracks (the "master recording") which can be reproduced by a typical playback system, such as a CD player or cassette deck.

7.12.2 Recording Studio Equipment

In modern recording studios many traditional components are being replicated with computer technology. The essential equipment found in most recording studios is as follows:

1. **Computer**

Computers are a central component of modern recording studios. With a computer, you can record and mix music using a digital audio work station such as Pro Tools, Cubase, Sonar, or Logic Pro, as well as use a variety of software synthesizers and effects.

2. **Audio Interface**

Audio interfaces allow you to connect audio devices to the computer. The ones designed for recording typically have many audio inputs for microphones and line level instruments, audio outputs for studio monitors and headphones and MIDI inputs and outputs. In most cases, they can connect to the computer via a USB or IEEE 1394 (FireWire) cable.

3. **Studio Monitors and Headphones**

Studio monitors are loudspeakers designed to reflect source audio as accurately as possible. They are typically used by producers and sound engineers to monitor audio during recording, as well as to play back audio. Similarly, headphones are necessary for musicians to be able to hear background audio such as click tracks and other instruments while recording.

4. **Microphones**

Microphones convert sound waves to electrical impulses. When plugged into a computer via an audio interface, these impulses become digitized and can be recorded. Recording studios often have many different microphones for recording various types of sounds. For example, some microphones are designed specifically to capture vocals, while others are designed to capture instruments.

5. **Rack Effects**

Rack effects apply one or more filters to audio signals to change the way they sound. Although computers can produce nearly every type of audio effect today, rack effects, particularly vintage ones, are still commonly used in professional recording studios.

6. **Controllers**

Controllers are external devices used to control computer software. The most common type of controller is a MIDI controller, which has a keyboard much like a standard electronic keyboard, although it does not actually produce audio like an electronic keyboard.

7. **DI Boxes**

Direct input (DI) boxes convert line level signals to balanced signals. They are often used to plug electric guitars and bass guitars into an XLR (the type of connection used by most microphones) input. They are only necessary if the audio interface does not have line level inputs.

8. Cables

Cables are an important part of any recording studio. XLR cables are commonly used to connect microphones to audio interfaces, while 1/4-inch cables are commonly used to patch other devices together.

9. Miscellaneous Items

A number of miscellaneous items are often found in recording studios, including microphone stands and shockmounts, power conditioners, furniture, soundproofing materials, vintage gear such as tube microphone preamps and a collection of musical instruments.

7.13 Home Theater System

Home theater system is a combination of electronic components designed to recreate the experience of watching a movie in a theater. When we watch a movie on a home theater system, it gives a sense of good experience than watching on an ordinary television.

To build a home theater, we need to create the following elements.

- A large screen television (32 inches) with a clear picture.
- Atleast four speakers.
- Equipment for splitting up the surround sound signal and sending it to the speakers.
- The main thing that sets a home theater, which differs from an ordinary television setup, is the surround sound. For a proper surround sound system, two or three speakers in front of the viewer. The audio signal is split into multiple channels so that different sound information comes out of the various speakers. The most prominent sound comes out of the front speakers. When someone or something is making noise on the left side of the screen, we hear it more from a speaker to the left side. Similarly in the right side, we hear

from the right side speaker. The third speaker sits in the center, just under or above the screen. This center speaker is very important because it anchors the sound coming from the left and right speakers. It plays all the dialogue and front sound effects so that they seem to be coming from the center of the television screen rather than from the sides.

The speakers behind the viewer fill in various sorts of background noise in the movie such as dog's barking, rushing water and the sound of a plane overhead. They also work with the speakers in front of the viewer to give the sensation of the movement. A sound starts from the front and then moves behind the viewer.

7.14 Noise pollution

Noise pollution is a type of energy pollution in which distracting, irritating or damaging sounds are freely audible. It is a dangerous pollutant, even destroys bridges and produces cracks in buildings. The noise can cause skin and mental diseases. The various sources of noise pollution are shown in Figure 7.18.

Pollution of Air by Sound

The intensity of sound is measured in decibels. The various ranges and sources of sound pollution are given in Table 7.1. All are responsible for the noise pollution because most of our day–to–day activities generate some noise. Often neglected, this pollution adversely affects the human beings leading to irritation, loss of concentration, loss of hearing and many more.

From early morning, we hear the horns of vehicles like trucks, buses, scooters and motor cycles. The drivers always use the horns more out of habit than necessity. On a special day like festivals, marriage functions, birthday parties and from religious places, we can hear loud speakers sound drilling the common man with severe noise pollution.

Adverse Effects of Noise Pollution

Noise effect is harmful to human beings, environment and animals in many ways. Some of them are as follows.

1. Hearing Problems

Exposure to noise can damage one of the most vital organs of the body, the ear. Hearing impairment due to noise pollution can either be temporary or permanent. When the sound level crosses the 70 decibel (dB) mark, it becomes noise, for the ear. Above 80 dB produces damaging effects to the ear.

When ear is exposed to extreme loud noise, above 100 dB for a considerable period of time, it can cause irreparable damage and may lead to permanent hearing loss.

2. Cardiovascular issue

A noisy environment can be a source of heart related problems. High intensity sound causes a dramatic rise in blood pressure as noise levels constrict the arteries, disrupting the blood flow. The heart rate also increases and become one of the reasons to the cardiovascular diseases.

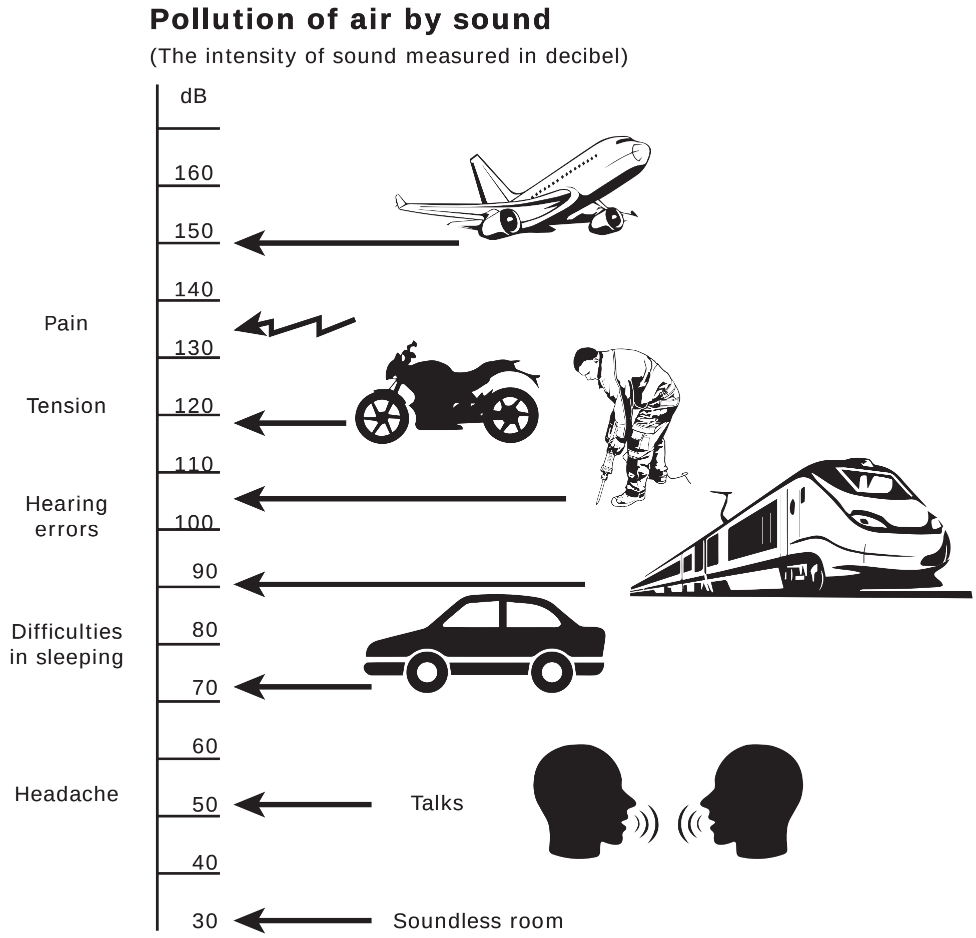


FIGURE 7.17 Various sources of sound pollution and their effects

TABLE 7.1. Intensity of sound noise sources and human perception

Noise source	Intensity of sound (dB)	Human perception
Threshold of hearing	0	Threshold of hearing
Breathing	10	Just audible
Sound of leaves in trees	20	Very quiet
Whispering	30	Very quiet
Normal conversation	30-40	Quiet
Homes and Restaurant	45-50	Quiet
Loud conversation	65	Moderately loud
Lawn mower	60-80	Moderately loud
Vacuum cleaner	80	Moderately loud
Traffic noise	60-90	Loud
Heavy trucks	90-100	Very loud
Thunderstorm	110	Very loud
Rock music	120	Uncomfortably loud
Jet take off (100 m distance)	120	Uncomfortably loud
Jet engine (at 15 m distance)	140	Painfully loud
Rocket engine	170-180	Painfully loud

3. **Sleep disturbance**

This is one of the noise pollution effects that can deter humans overall well-being. Noise can interrupt good night sleep, when this occurs the person feels extremely annoyed and uncomfortable. The disturbed person's energy level fall down considerably and decreases the ability to work efficiently.

Control of Noise Pollution

Due to the various impacts of noise on human beings and environment, it should be controlled. There are four fundamental ways in which noise can be controlled. They are

1. Reduce noise at the source.
2. Block the path of the noise.
3. Increase the path length of noise.
4. Protect the recipient.

7.15 Government Rules and Regulations Regarding Limit Sound

The central pollution control board of India published a rule book, with title the noise pollution (regulation and control) rules , 2000' in the year of 2000.

In this rule book, they have devided all areas in four different zones and decided limits for noise level in respective zones.

1. industrial area : 75 dBA (Day time)
   70 dBA (Night time)
2. commercial area : 65 dBA (Day time)
   55 dBA (Night time)
3. residential area : 55 dBA (Day time)
   45 dBA (Night time)
4 silence zone : 50 dBA (Day time)
   40 dBA (Night **time)**

Note

1. Day time shall mean from 6.00AM to 10.00PM
2. Night time shall mean from 10.00PM to 6.00AM
3. Silence zone is an area comprissing not less than 100 meters around hospitals, educational institutions, courts, religious places or any other area which is declared as such by the competent authority.
4. dBA (A – weighted decibel) is unit of noise

Some of the noise limits for vehicles depending upon the capacity of their engines.

- 77 dBA for two wheelers between 80cc to 175cc engines
- 75 dBA for two wheelers more than 175cc engines
- 75 dBA for cars (less than 9 seater)
- 80 dBA for heavy vehicles

These are the standard noise limit which have been accepted by government of India.

Loudspeakers may be used with the permission of relevant authority .

The Public Address System(PAS) cannot be used in the night time except in closed areas.

LEARNING OUTCOME

After completing this chapter, the students can understand the following

- **Characterices of sound waves**
- **Basics of acoustic engineering in auditorium**
- **Applications of Acoustic Engineering**
- **PA system and Audio power amplifier circuits**
- **DTS/DOLBY sound systems in theater**
- **How to control noise pollution**

GLOSSARY

Acoustic	The science and scientific study of sound. The properties of a room or environment that affect the qualities of sound.
Ambient noise level	'Background' noise-from any source that affects the listeners ability to hear what is produced by a sound systems. Machinery, hum from florescent lights, traffic etc.,
Attenuate	To make weaker. An attenuator uses resistance to reduce output voltage, as with a volume control.
Bass	The lower end of the frequency range, from about 20 Hz to 300 Hz.
dB (Decibel)	A relative unit of measure between two sounds or a radio signals. A difference 1 dB is considered to be the smallest that can be detected by the human ear. An increase of 6 dB equals twice the sound pressure. As a measure of sound pressure levels, used to indicate loudness.
Equaliser	A device that permits the precise control of specific frequency ranges. Examples are: Graphic, parametric, notch filter, cut only.
Filter	A device that removes unwanted frequencies or noise from a signal
Frequency	The number of times that a periodic function or vibration occurs or repeats itself in one second
Frequency response	The range of frequencies that are reproducible by a speaker or electronic component
Hz (Hertz)	A unit of measure that equals one cycles per second
Impedance	The measure of total resistance to the current flow in an alternating current circuit; expressed in ohms, as a characteristics of electrical devices (Speakers and microphone)

Mixer	An electronic device that permits the combining of a number of inputs into one or more outputs. Mixers commonly provider of variety of controls-tone, volume, balance and effects – for each 'channel'.
Pitch tone	A function of frequency
Reflection	A term that describes the amount of sound 'bouncing' off of hard surfaces.
Reverberation	Sound waves that confirm to bounce around a space after the sound source has ended
Room	Any a closed space in which a performance is stayed. It can be as small as a closed or as large as the super dome
SPL (Sound Pressure Level)	The measurement of the loudness or amplitude of sound expressed in decibels (dB)
Transducer	A device which converts sound into electrical energy (a microphone), or electrical energy into sound (a spectrum)

QUESTIONS

Part – A (1 Mark)

I Multiple choice Questions

1. Sound is produced due to _____

   a. Friction b. Circulation
   c. Vibration d. Refraction

2. Sound waves travel at _____

   a. Same speed in different mediums
   b. Different speed in same mediums
   c. Different speed in different mediums
   d. Highest speed in vacuum

3. Sound wave do not travel throguh ___________

   a. Vacuum b. Solid
   c. Liquid d. Gases

4. The wavelength of a wave is measured in _____

   a. Meters
   b. Hertz
   c. Seconds
   d. Decibels

5. Range of frequencies which human ear hear is called ______________

   a. Pitch of sound
   b. Loudness of sound
   c. Audible frequency of sound
   d. Quality of sound

6. Sounds above 20000 Hz is called _____

   a. Ultra cool b. Ultra sound
   c. Infra-audio d. Infrasound

7. The velocity of sound in air ____________

   a. 300 m/s b. 334 m/s
   c. 1130 m/s d. 350 m/s

8. Which microphone will be damaged if exposed to high temperature above 52^0C?

a. Dynamic b. Crystal
c. Rubber d. condenser

9. The part of the ear that responds to sound waves like a microphones diaphragm is the _____________
   a. Lobe
   b. Ear drum
   c. Bones of the middle ear
   d. Fluid in the ear

10. _____ is early reflection of sound
   a. Echo
   b. Reverberation
   c. Pure sound
   d. Intelligible sound

11. Sound which has jarring and unpleasant effect on our ears is called?
   a. Frequency b. Amplitude
   c. Noise d. Musical sounds

12. Multiple reflections are called _____________
   a. Reverberations
   b. Refraction
   c. Echo
   d. Compressions

13. A higher pitch means?
   a. Zero frequency
   b. Lower frequency
   c. Higher frequency
   d. Lower loudness

14. Bass response is
   a. Maximum high frequency response
   b. Emphasizing the high audio frequency
   c. By passing high audio frequency
   d. By passing low audio frequency

15. The noise level of industrial area decided by the central Government is between _____________
   a. 40-50 dBA
   b. 45-55 dBA
   c. 55-65 dBA
   d. 70-95 dBA

Part – B (3 Marks)

II Answer in few sentences

1. What is sound? How it is produced?
2. Differentiate low and high pitch?
3. In which medium, the sound can travel in higher speed? What its velocity?
4. Write the advantages of Headphones
5. Name few characteristics of sound waves
6. What is the function of mixer stage in PA system
7. Why noise pollution is dangerous?
8. What are the techniques used to control noise pollution
9. What is stereo effect?
10. Name few equipments needed to build a home theater

Part – C (5 Marks)

III Answer in a paragraph

1. Explain the two types of sound waves
2. Write short notes on stereophonic effects
3. Explain the working functions of crystal microphone
4. Draw and explain audio amplifier using TDA 2003.

Part – D (10 Marks)

IV Answer in One Page (Essay type Question)

1. Explain the various applications of Acoustic Engineering
2. How acoustic engineering is essential in auditorium and theater design? Explain in detail.
3. Draw the block diagram of PA system and explain each block
4. Explain DTS & Dolby techniques in theater sound system.

Answers

1. (c)	2. (c)	3. (a)	4. (a)
5. (c)	6. (b)	7. (b)	8. (b)
9. (b)	10. (a)	11. (c)	12. (a)
13. (c)	14. (c)	15. (d)	